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**BEFORE THE BOARD OF PATENT APPEALS
AND INTERFERENCES**

Application Number: 10/731,929
Filing Date: December 10, 2003
Appellant(s): KEMP ET AL.

Bradley D. Lytle
Reg. No. 40,073

Nikolaus P. Schibli
56,994
For Appellant

EXAMINER'S ANSWER

This is in response to the appeal brief filed 17 August 2009 appealing from the Office action mailed 22 October 2008.

(1) Real Party in Interest

A statement identifying by name the real party in interest is contained in the brief.

(2) Related Appeals and Interferences

The examiner is not aware of any related appeals, interferences, or judicial proceedings which will directly affect or be directly affected by or have a bearing on the Board's decision in the pending appeal.

(3) Status of Claims

The statement of the status of claims contained in the brief is correct.

(4) Status of Amendments After Final

The appellant's statement of the status of amendments after final rejection contained in the brief is correct.

(5) Summary of Claimed Subject Matter

The summary of claimed subject matter contained in the brief is correct.

(6) Grounds of Rejection to be Reviewed on Appeal

The appellant's statement of the grounds of rejection to be reviewed on appeal is correct.

(7) Claims Appendix

The copy of the appealed claims contained in the Appendix to the brief is correct.

(8) Evidence Relied Upon

2005/0060153	Gable et al.	3-2005

Brandstein, M. S.. et al. "Microphone-Array Localization Error Estimation with Application to Sensor Placement," June 1996, Journal of the Acoustical Society of America 99(6), p 3807-3816

Lee, C. M. et al., "Recognition of Negative Emotions from a Speech Signal," 9-13 Dec 2001, ASRU '01 IEEE Workshop on Automatic Speech Recognition and Understanding, pages 240-243.

(9) Grounds of Rejection

The following ground(s) of rejection are applicable to the appealed claims:

Claim Rejections - 35 USC § 103

1. Claims 1, 2, 4-9, and 12-14 rejected under 35 U.S.C. 103(a) as being unpatentable over Gable et al. (US PAP 2005/0060153) in view of Brandstein et al. (Microphone Array Localization Error Estimation with Application to sensor Placement).

2. Consider claim 1, Gable teaches a method for processing speech (abstract), comprising the steps of:

receiving a speech input of a speaker (figure 2, microphone 202; paragraph 0030),

generating speech parameters from said speech input (parameters extracted; paragraph 0027),

determining parameters describing an absolute loudness of said speech input (amplitude is determined; paragraph 0027),

evaluating said speech input and/or said speech parameters using said parameters describing the absolute loudness to identify the speaker (paragraph 0027-0029, parameters evaluated to verify a user identity.).

Gable does not specifically teach the absolute loudness being a loudness of the speech at a location of a source of the speech.

In the same field of speech processing, Brandstein suggests the absolute loudness being a loudness of the speech at a location of a source of the speech (Section 2 discusses using a microphone array with a time difference of arrival algorithm to determine a location of a speaker; pages 3-5. Page 21 teaches modeling a source as a cardioid radiator, wherein the source amplitude is a function of distance from the source. When this information is combined with the source locating algorithms of section 2, one can obviously estimate the amplitude at the source itself given the amplitude at the microphone array).

Therefore it would have been obvious to one of ordinary skill in the art at the time if the invention to combine the absolute loudness as suggested by Brandstein with the speech system of Gable in order to provide a method of normalizing the loudness for

speaker verification to provide a means for provide a high quality signal of the desired speaker that is not adversely effected by the distance from a speaker to the microphone array. (Introduction, Brandstein.).

3. Consider claim 4, Gable does not, but Brandstein teaches a method according to claim 1 wherein a microphone array comprising a plurality of microphones (see figure 6) is used for determining said parameters describing the absolute loudness (Existing array systems have been used in a number of applications. These include teleconferencing, speech recognition, speaker identification, speech acquisition in an automobile environment, sound capture in reverberant enclosures, large room recordings, conferencing, acoustic surveillance, and hearing aid devices; page 1 lines 11-15. Obviously, the array of microphones would be used to determine the parameters including loudness needed for these applications.).

Therefore it would have been obvious to one of ordinary skill in the art at the time if the invention to combine the multiple microphones as suggested by Brandstein with the speech system of Gable in order to provide a means of normalizing the loudness for speaker verification to provide a means for provide a high quality signal of the desired speaker that is not adversely effected by the distance from a speaker to the microphone array. (Introduction, Brandstein.).

4. Consider claim 5, Gable does not, but Brandstein teaches a method according to claim 1 wherein a location and/or distance of the speaker is determined (Section 2

discusses using a microphone array with a time difference of arrival algorithm to determine a location of a speaker; pages 3-5.).

Therefore it would have been obvious to one of ordinary skill in the art at the time if the invention to combine the localization as suggested by Brandstein with the speech system of Gable in order to provide a means of normalizing the loudness for speaker verification to provide a means for provide a high quality signal of the desired speaker that is not adversely effected by the distance from a speaker to the microphone array. (Introduction, Brandstein.).

5. Consider claim 6, Gable does not, but Brandstein teaches a method according to claim 1 wherein the absolute loudness is determined using algorithms for auditory and/or binaural processing (Page 21 teaches modeling a source as a cardioid radiator, wherein the source amplitude is a function of distance from the source. When this information is combined with the source locating algorithms of section 2, one can obviously estimate the amplitude at the source itself given the amplitude at the microphone array.).

Therefore it would have been obvious to one of ordinary skill in the art at the time if the invention to combine the source model as suggested by Brandstein with the speech system of Gable in order to provide a means of normalizing the loudness for speaker verification to provide a means for provide a high quality signal of the desired speaker that is not adversely effected by the distance from a speaker to the microphone array. (Introduction, Brandstein.).

6. Consider claim 7, Gable does not, but Brandstein teaches a method according to claim 5, wherein said absolute loudness is computed by normalizing a measured loudness, or energy by said distance (Page 21 provides a relationship of a source amplitude as a function of distance and angle form the source. This relationship could obviously be used to normalize an amplitude value to estimate the amplitude at the source.).

Therefore it would have been obvious to one of ordinary skill in the art at the time if the invention to combine the distance normaliztion as suggested by Brandstein with the speech system of Gable in order to provide a means of normalizing the loudness for speaker verification to provide a means for provide a high quality signal of the desired speaker that is not adversely effected by the distance from a speaker to the microphone array. (Introduction, Brandstein.).

7. Consider claim 8, Gable does not, but Brandstein teaches a method according to claim 5, wherein said distance is determined using the time delay of the speech input between said plurality of microphones (Sections 2 and 3 discuss using a microphone array with a time difference of arrival algorithm to determine a location of a speaker; pages 3-10.)

Therefore it would have been obvious to one of ordinary skill in the art at the time if the invention to combine the multiple microphones as suggested by Brandstein with the speech system of Gable in order to provide a means of normalizing the loudness for

speaker verification to provide a means for provide a high quality signal of the desired speaker that is not adversely effected by the distance from a speaker to the microphone array. (Introduction, Brandstein.).

Consider claim 9, Gable teaches a speech processing system (abstract), configured to:

receive a speech input of a speaker (figure 2, microphone 202; paragraph 0030),

generate speech parameters from said speech input (parameters extracted; paragraph 0027),

determine parameters describing an absolute loudness of said speech input (amplitude is determined; paragraph 0027),

evaluate said speech input and/or said speech parameters using said parameters describing the absolute loudness to identify the speaker (paragraph 0027-0029, parameters evaluated to verify a user identity.).

Gable does not specifically teach the absolute loudness being a loudness of the speech at a location of a source of the speech.

In the same field of speech processing, Brandstein suggests the absolute loudness being a loudness of the speech at a location of a source of the speech (Section 2 discusses using a microphone array with a time difference of arrival algorithm to determine a location of a speaker; pages 3-5. Page 21 teaches modeling a source as a cardioid radiator, wherein the source amplitude is a function of distance from the source. When this information is combined with the source locating algorithms of

section 2, one can obviously estimate the amplitude at the source itself given the amplitude at the microphone array).

Therefore it would have been obvious to one of ordinary skill in the art at the time if the invention to combine the absolute loudness as suggested by Brandstein with the speech system of Gable in order to provide a method of normalizing the loudness for speaker detection to provide a means for provide a high quality signal of the desired speaker that is not adversely effected by the distance from a speaker to the microphone array. (Introduction, Brandstein.).

8. Consider claim 12, Gable teaches a computer readable medium encoded with a computer program configure to cause a processor based device to execute the method of (Abstract, computer figure 2 requires a program):

receiving a speech input of a speaker (figure 2, microphone 202; paragraph 0030),

generating speech parameters from said speech input (parameters extracted; paragraph 0027),

determining parameters describing an absolute loudness of said speech input (amplitude is determined; paragraph 0027),

evaluating said speech input and/or said speech parameters using said parameters describing the absolute loudness to identify the speaker (paragraph 0027-0029, parameters evaluated to verify a user identity.).

Gable does not specifically teach the absolute loudness being a loudness of the speech at a location of a source of the speech.

In the same field of speech processing, Brandstein suggests the absolute loudness being a loudness of the speech at a location of a source of the speech (Section 2 discusses using a microphone array with a time difference of arrival algorithm to determine a location of a speaker; pages 3-5. Page 21 teaches modeling a source as a cardioid radiator, wherein the source amplitude is a function of distance from the source. When this information is combined with the source locating algorithms of section 2, one can obviously estimate the amplitude at the source itself given the amplitude at the microphone array).

Therefore it would have been obvious to one of ordinary skill in the art at the time if the invention to combine the absolute loudness as suggested by Brandstein with the speech system of Gable in order to provide a method of normalizing the loudness for speaker detection to provide a means for provide a high quality signal of the desired speaker that is not adversely effected by the distance from a speaker to the microphone array. (Introduction, Brandstein.).

9. Consider claim 13, Gable teaches a method for processing speech (Abstract, computer figure 2 requires a program), comprising:

receiving a speech signal of a speaker (figure 2, microphone 202; paragraph 0030);

generating speech parameters from said speech signal (parameters extracted; paragraph 0027); and

evaluating at least one of said speech signal and said speech parameters using the normalized loudness or energy to identify the speaker (paragraph 0027-0029, parameters evaluated to verify a user identity.).

However Gable does not specifically teach:

determining a distance of the speaker based on a time delay of a respective arrival of said speech signal at two or more microphones; and

normalizing a measured loudness or energy by said distance, and

calculating an absolute loudness being a loudness of a speech that generated the speech signal at a location of a source of the speech.

In the same field of speech processing, Brandstein teaches determining a distance of the speaker based on a time delay of a respective arrival of said speech signal at two or more microphones (Sections 2 and 3 discuss using a microphone array with a time difference of arrival algorithm to determine a location of a speaker; pages 3-10.); and

normalizing a measured loudness or energy by said distance (Page 21 provides a relationship of a source amplitude as a function of distance and angle form the source. Although this relationship was given to model the source, one of ordinary skill in the art at the time of the invention would have thought, given the location of the source (as determined in the localization method discussed throughout Brandstein) and

the detected amplitude at the microphone array, to use the relationship to determine the source amplitude) and

calculating an absolute loudness being a loudness of a speech that generated the speech signal at a location of a source of the speech (Section 2 discusses using a microphone array with a time difference of arrival algorithm to determine a location of a speaker; pages 3-5. Page 21 teaches modeling a source as a cardioid radiator, wherein the source amplitude is a function of distance from the source. When this information is combined with the source locating algorithms of section 2, one can obviously estimate the amplitude at the source itself given the amplitude at the microphone array).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to use a microphone array for source location and absolute volume as suggested by Brandstein with the speech processing system of Gable in order to provide a means for provide a high quality signal of the desired speaker that is not adversely effected by the distance from a speaker to the microphone array. (Introduction, Brandstein.).

10. Consider claim 14, Gable teaches a system for emotion recognition and/or speaker identification, comprising:

a data processor configured to generate speech parameters from said speech signal (parameters extracted; paragraph 0027), and

further configured to evaluate at least one of said speech signal and said speech parameters using the normalized loudness or energy to identify the speaker (paragraph 0027-0029, parameters including amplitude evaluated to verify a user identity.).

However Gable does not specifically teach:

at least two microphones configured to receive a speech signal; and

a processor configured to determine a distance of the speaker based on a time delay of a respective arrival of said speech signal at said microphone, to normalize a measured loudness or energy by said distance and calculating an absolute loudness being a loudness of a speech that generated the speech signal at a location of a source of the speech

In the same field of speech processing Brandstein teaches:

at least two microphones configured to receive a speech signal (see microphone array in figure 6); and

a processor configured to determine a distance of the speaker based on a time delay of a respective arrival of said speech signal at said microphone (Sections 2 and 3 discuss using a microphone array with a time difference of arrival algorithm to determine a location of a speaker; pages 3-10.), to normalize a measured loudness or energy by said distance (Page 21 provides a relationship of a source amplitude as a function of distance and angle form the source. Although this relationship was given to model the source, one of ordinary skill in the art at the time of the invention would have thought, given the location of the source (as determined in the localization method discussed throughout Brandstein) and the detected amplitude at the microphone array, to use the

relationship to determine the source amplitude) and calculating an absolute loudness being a loudness of a speech that generated the speech signal at a location of a source of the speech (Section 2 discusses using a microphone array with a time difference of arrival algorithm to determine a location of a speaker; pages 3-5. Page 21 teaches modeling a source as a cardioid radiator, wherein the source amplitude is a function of distance from the source. When this information is combined with the source locating algorithms of section 2, one can obviously estimate the amplitude at the source itself given the amplitude at the microphone array).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to use a microphone array for source location and absolute volume as suggested by Brandstein with the speech processing system of Gable in order to provide a means for provide a high quality signal of the desired speaker that is not adversely effected by the distance from a speaker to the microphone array. (Introduction, Brandstein.).

11. Consider claim 15, Gable teaches a method for processing speech (abstract) comprising the steps of:

receiving a speech signal of a speaker (figure 2, microphone 202; paragraph 0030);

calculating an absolute loudness (amplitude is determined; paragraph 0027);

determining features from the speech signal, wherein the features are at least partly based on the absolute loudness (amplitude is determined; paragraph 0027); and

determining an identity of the speaker based on the features (paragraph 0027-0029, parameters evaluated to verify a user identity.).

Gable does not specifically teach the absolute loudness being a loudness of the speech at a location of a source of the speech.

In the same field of speech processing, Brandstein suggests the absolute loudness being a loudness of the speech at a location of a source of the speech (Section 2 discusses using a microphone array with a time difference of arrival algorithm to determine a location of a speaker; pages 3-5. Page 21 teaches modeling a source as a cardioid radiator, wherein the source amplitude is a function of distance from the source. When this information is combined with the source locating algorithms of section 2, one can obviously estimate the amplitude at the source itself given the amplitude at the microphone array).

Therefore it would have been obvious to one of ordinary skill in the art at the time if the invention to combine the absolute loudness as suggested by Brandstein with the speech system of Gable in order to provide a method of normalizing the loudness for emotion detection to provide a means for provide a high quality signal of the desired speaker that is not adversely effected by the distance from a speaker to the microphone array. (Introduction, Brandstein.).

12. Claim 2 is rejected under 35 U.S.C. 103(a) as being unpatentable over Gable and Brandstein as applied to claim 1 above, and further in view of Lee et al. (Recognition of Negative Emotions from the Speech Signal).

13. Consider claim 2, Gable and Brandstein teaches a method according to claim 1, but does not specifically teach wherein the step of evaluation comprises a step of emotion recognition.

In the same field of speech processing, Lee teaches a step of emotion recognition (This paper reports on methods for automatic classification of spoken utterances based on the emotional state of the speaker; page 240, column 2, lines 3-4.).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to combine the emotion detection of Lee, with the system of Gables and Brandstein in order to properly classify emotional information that may affect speaker recognition and verification.

(10) Response to Argument

i) The Applied Prior Art References, Taken in any Proper Combination, Do Not Teach all the Features of Appellant' Independent Claim 13

1. The Appellant asserts on pages 8 and 9:

> But Brandstein is silent on a step of calculating an absolute loudness being a loudness of a speech that generated the speech signal at a location of a source of the speech, as required by Appellants' independent Claim 13...
> Appellants disagree with these assertions of the final Office Action on pages 11-12. Although the mathematical model for the source of Brandstein explains that a source amplitude can depend on radiation angle and distance, (Brandstein, p. 21, 11.10-14) Brandstein never actually calculates a loudness of a speech that generated by the speech signal at a location of a source of the speech, as required in Appellants' Claim 13. Brandstein explains on pages 3-4 that unbiased estimates of time-difference-of-arrival (TDOA) of acoustic signals

> are calculated using propagation speeds and a maximum-likelihood estimation algorithm (Brandstein, from p. 4, 1.4, to p. 5, 1.3.) Similarly, in Section 3.2, related to the source estimate, Brandstein uses maximum-likelihood algorithms to calculate a geometric location of a source, using the estimated TDOAs. (Brandstein, pp. 8-10. Equation 16.) But Brandstein never calculates a source amplitude, nor does he explain how the amplitude can be calculated by giving a simple model thereof.

The Examiner maintains that one of ordinary skill in the art could use Brandstein to determine an amplitude at a source, thus normalizing the amplitude detected at a microphone array by the distance from the source. As pointed out the Appellant, sections 4.1.4 to 5.1.3 shows how one can locate a source using time-difference-of arrival (TDOA) between different microphones in an array of microphones. Using this technique with received signal amplitude, one can calculate the position of a sound source, and thus, the distance from the source to the microphone arrays.

The Appellant contends that it would not have been to use the source model disclosed in section 5.1, page 21 to estimate the amplitude of the sound at the source itself. The Examiner disagrees. Brandstein discloses a mathematical proportion that relates the perceived amplitude at a distance to the distance and angle. Using this proportion, one with a background in acoustics would be able to develop an equation to use for calculating an amplitude at a source, given the distance, angle, and amplitude detected at a microphone. So, determining source amplitude is a matter of mere mathematical manipulations. Even when other variables are considered, the proportion described by Brandstein still applies. It is the examiners position, that it would have been obvious to one of ordinary skill in the art to use this proportion to estimate the amplitude at the source itself, taking into account other variables as precision requires.

2. The Appellant asserts on page 9:

> Regarding the Office Action's assertion that "one can obviously estimate the amplitude at the source itself given the amplitude at the microphone array," Appellants believe that this assertion of inherency of the features of Appellants' Claim 13 is insufficient to reject this claim. A mere position that a reference could perform a claimed feature is insufficient to form a rejection based on inherency.

The examiner points out that no arguments were ever made by the examiner based on inherency. The Examiner contended that although Brandstein did not specifically teach determining a distance normalized amplitude, Brandstein provided all the necessary tools to do so. The examiner contention that, as explained above, it was a matter of mere mathematical manipulation to determine a source amplitude by "unnormalizing" the normalized amplitude, and that the manipulations would have been obvious to one of ordinary skill in the art of acoustics.

ii) The Combination of the Applied Prior Art References is Not Obvious

3. The Appellant asserts on pages 11 and 12:

> First, from the above reasoning, it is still not clear how Brandstein's multi-speaker system with four microphones installed in a room, to identify a location of a targeted speech source, (Brandstein, Fig. 6, "microphone placements") could be incorporated into Gable's speaker verification system, having a single microphone 202 and a GEMS sensor 208 that are connected to a PC 210. (Gable, Fig. 2, p. 2, ¶ [0030].) Under such a modification, Gable's system could not be used as suggested in the pending Office Action, because in Gable's system, only one microphone 202 is used, and the speaker must approach this single microphone to make "an identity claim" during the speaker identification process 100. (Gable, p. 2, ¶ [0029].) In addition, Gable's verification system is

> designed to heavily rely on information from a non-acoustic sensor 208. (Gable, p. 2, ¶ [0026], Fig. 2).
>
> Therefore, the introduction of a multi-microphone system in a room with many speakers would clearly require a substantial reconstruction or redesign of the elements of Gable where only one microphone 202 and a GEMS sensor 208 is used that is approached by a person to make a identification statement, and such redesign would change the basic principle of operation of Gable. There is no evidence that a person of ordinary skill in the art would be motivated to perform such changes and redesign…
>
> Second, in order to form the combination of a reference with the teachings of Gable and Brandstein, one of ordinary skill in the art would look for a reference directed to voice identification, and the estimation or calculation of absolute loudness, as required i.e. by Appellants' Claim 13.
>
> But Gable makes it clear his system heavily relies on the information delivered by the non-acoustic GEMS sensor 208, and Brandstein is only proposing a solution to geographically locate one speaker in a room with many users of a video conferencing system based on acoustic data. (Gable, p. 2, ¶ [0026], 11.14-15, "It]he EM data also provides that were previously unobtainable with the all-acoustic verification systems"). Therefore, the reference Brandstein is directed to a different field of application and a different problem, but also does not teach anything related to the calculation of an absolute loudness, as discussed above with respect to subparagraph i). One of ordinary skill in the art would, therefore, not look out to the reference Brandstein to combine it with the reference Gable.

The examiner contends that Brandstein could be combined with Gable by one of ordinary skill in the art without significantly changing the operation of Gable's speaker verification. Brandstein points out in the introduction, that microphone arrays are commonly used in systems that include speaker identification systems. Thus it would have been obvious to substitute for a single microphone Gable the array found in Brandstein, as this is already common practice. By using an array instead of single microphone, one would eliminate the need in Gable to approach the microphone, one could just be relatively close to the array because of the directional filtering by the array. The localization techniques taught by Brandstein would compensate. The GEM

sensors in Gable are used mostly for pitch estimation (paragraphs 0034 and 0035) not for amplitude estimation. Thus the microphone array is relied upon for amplitude extraction.

A problem with microphones recognized by those of ordinary skill in acoustics is that the further a source is from a microphone, the lower the detected amplitude is. Thus there exists an obvious need to “normalize” the amplitude by the distance in order to provide accurate identification. As discussed above, Brandstein provides the tools to locate a source in space, thus determining the distance and angle. Brandstein further provides a source model which can, through simple mathematical manipulation, but used to determine the amplitude of a source at the source itself. A third reference is not needed, because the answer would be found in mathematical manipulations of the proportions disclosed by Brandstein.

iii) Appellants Independent Claims 1, 9, 12, and 14-15 are also Believed to be Patently Distinct Over the Applied Prior Art References.

4. The Appellant asserts on page 13:

> Because independent Claims 1, 9, 12, and 14-15 include features that are analogous to the features argued above in subparagraph i) with respect to the "absolute loudness," and Claims 1, 9, 12, and 14-15 have been rejected based on an analogous obviousness rejection over the references Brandstein and Gable, Appellants respectfully submit that the arguments presented above in subparagraphs i) and ii) towards patentability of independent Claim 13 are also applicable to the patentability of independent Claims 1, 9, 12, and 14-15.

The examiner maintains that these rejections are proper for the reasons listed in the discussion of sections i) and ii).

iv) Appellants' Dependent Claim 2 is Believed to Be Patentably Distinct Over the Applied Prior Art Reference.

5. The Apellant asserts on page 14:

> From the above discussion it is evident that the cited passages of Lee fail to teach a step of calculating an absolute loudness being a loudness of a speech that generated the speech signal at a location of a source of the speech, as required by Appellants' Claim 13. First, Lee uses a single microphone of a telephone for the recording, and second, Lee applies a normalization filter to all the samples. In addition, Lee clearly cites that the energy level of the speech signal as received at the microphone is calculated. Therefore, it is not possible that Lee is calculating an absolute loudness being a loudness of a speech that generated the speech signal at a location of a source of the speech, as required by Appellants' Claim 13. Therefore, Lee fails to remedy the above argued deficiencies in subparagraphs i), ii) and iii) of Gable and/or Brandstein, even if we assume that the combination is proper. Therefore, dependent Claim 2 is also believed to be allowable by virtue of their claim dependency from independent Claim 1.

The examiner maintains that Brandstein was relied upon to teach calculating an absolute loudness, not Lee. When Lee is combined with Gable Brandstein, the need for the normalization described in Lee would be elimination by the normalization that can be easily derived from teachings of Brandstein. Because Brandstein teaches the limitations as described in the discussion of section i) and ii), the examiner believes the combination of Gable, Branstein and Lee teaches the limitations of claim 2.

(11) Related Proceeding(s) Appendix

No decision rendered by a court or the Board is identified by the examiner in the Related Appeals and Interferences section of this examiner's answer.

For the above reasons, it is believed that the rejections should be sustained.

Respectfully submitted,

Douglas C. Godbold

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